EZ-Phone
(HSS SIP UA)

User's Manual

HSS-BLR-SDC-ICP-0048

Revision: 1.0

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Table of Contents

1. INTRODUCTION 4

2. FEATURES 4

3. TAB CONTROLS 5

A) KEYPAD 5
B) ONLINE BUDDY TAB 6

4. BUTTON CONTROLS 7

MAKE A CALL 7
HANG UP 8
AUDIO CONTROL 8
DO NOT DISTURB 8
CALL HOLD 8

5. MENU CONTROLS 9

FILE 9
C) EDIT 10
D) VIEW 14
E) HELP 14

6. STATUS MESSAGES 16

7. ERROR MESSAGES 17

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User Manual
For
HSS EZ-Phone Ver 1.0

1. Introduction

The Session Initiation Protocol (SIP) is an application-layer control protocol that can establish, modify and terminate multimedia sessions or calls. These multimedia sessions include multimedia conferences, distance learning, Internet telephony and similar applications. SIP can invite both persons and "robots", such as a media storage service.

The HSS EZ-Phone is a SIP client for Windows NT built upon the SIP stack of Hughes Software Systems.

2. Features

EZ-Phone (Evaluation Version) supports the following features:

- UDP support
- Call Hold
- Call Forward all calls
- Call Forward when busy
- Do Not disturb
- Choice of GSM, ADPMCM and u-Law codecs
- Buddy List

Note: This evaluation version of EZ-Phone uses SpeakFreely for transmitting/receiving media.

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3. Tab controls

The following are the tab windows present in the client.

Keypad

Online Buddy List

a) Keypad

The keypad provides a mouse-click interface for entering the SIP URL in the textbox. Presently it supports only digits from 0-9, "." and ":". The client automatically adds the "sip:" prefix if it is not provided.

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b) Online Buddy Tab

This shows the list of buddies. A green bullet next to the name indicates those buddies who are online (Online means that the remote user's SIP client is up). Similarly, a red bullet is used to indicate buddies who are offline.

One could keep a list of 'buddies' to call, instead of remembering the IP every time. New buddies can be added using the "Add New Friend" option.
For example, if one want to create a buddy called 'joe', we need to:

- Go to Edit | Add New Friend.
- In the dialog box, enter IP/host , port, and the name of the user (in this case Joe)
- Check the Add to Buddy List check box if one wants to show this in the buddy list (If not checked, then it will be a part of the address book, but not buddy list)

Now, one can click on the buddy list cue tab, and double click on the name to call.

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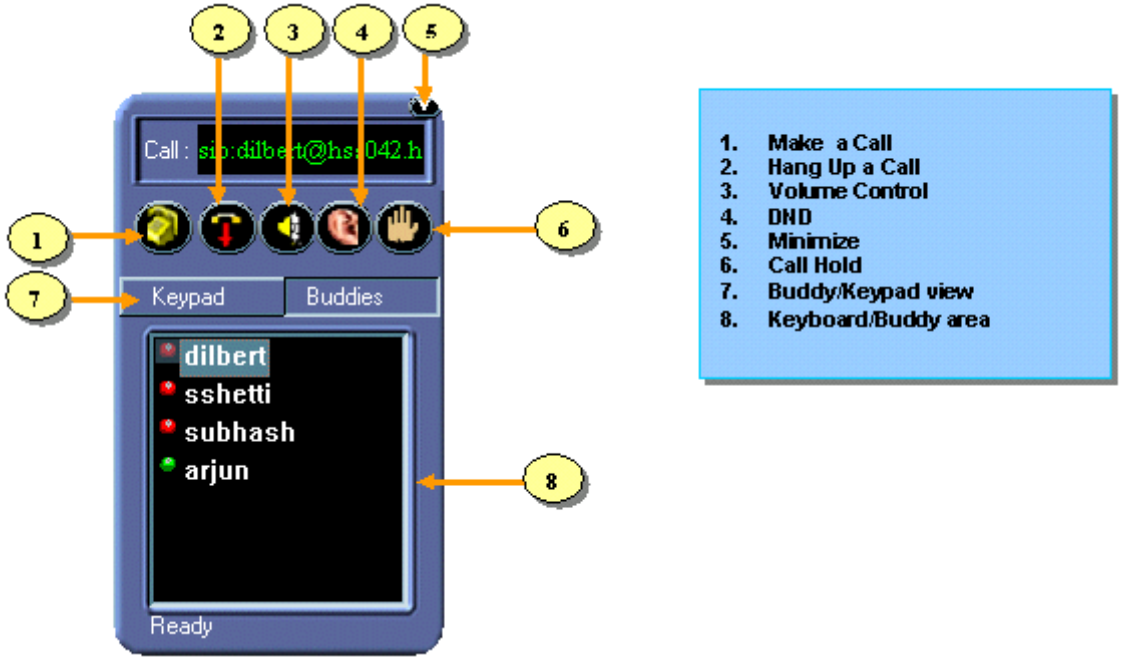
4. Button Controls

When a user places the mouse pointer on one of the buttons, tool tip for that button is shown. The following lists the buttons' tool tips and functionality it provides to the user.



Make a Call

The User can call any other remote client which supports the SIP protocol by entering the remote client's SIP URL in the text box provided (through the keypad on one of the tab controls or through the keyboard) and clicking on this button. (Note that a call can also be initiated by entering the URL and hitting "enter" in the Call box.

For example, one could type any of:

sip:myfriend@company.com:port

OR

company.com:port

Note that if no port is specified a default of 5060 is taken

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Hang up

The User can terminate an already established call by clicking on this button. In case the user is currently in two calls (using the call hold feature), clicking this button brings up a dialog giving the user the option to hang up one of the people he is connected to.
Also, the user can reset his client at any time (when he is not in a call-established state) by clicking this button.

Audio control

The User can adjust the preferred and playback settings by clicking on this button.

Do not Disturb

This is a toggle button. On clicking this button, the icon changes to an ear with a red cross. In this mode, the client will automatically reject incoming calls. Press again to disable.

Call Hold

This button is provided to the user so that he can switch between two calls. It works as follows:
The user has an established call with two remote clients in place. The calls may have been initiated by other remote clients or by the user himself. The user can alternately switch between these two established calls by clicking the Call Hold button. The client currently supports only two simultaneous calls.

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5. Menu Controls

The Menu bar appears in a separate window along with the EZ-Phone. This Menu Dialog can be closed by clicking the close button in the title bar. To access the Menu Dialog again just right click anywhere on the EZ-Phone. The following describes the functionality of the various menu options provided:

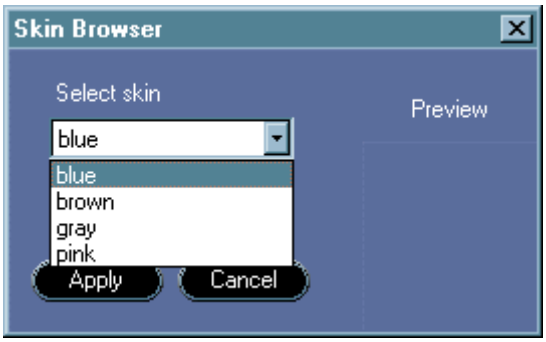
Menu Dialog

File

i) Register

This feature is not available in the evaluation release.

ii) Skin Browser



Use this option to select different skins for the client. The user could add any other skins by simply creating a new directory and inserting the new bitmaps with the following naming convention:

main.bmp	– skin for the main client
volctrl.bmp	– skin for the volume control
common.bmp	– skin for the rest of the dialogs.

iii) Reset

Use this option to reset the client.

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iv) Quit

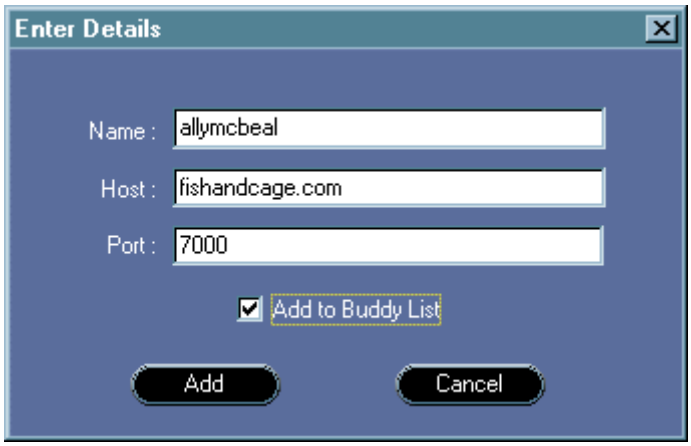
When this menu item is selected, it exits the SIP client.

c) Edit

i) Address Book

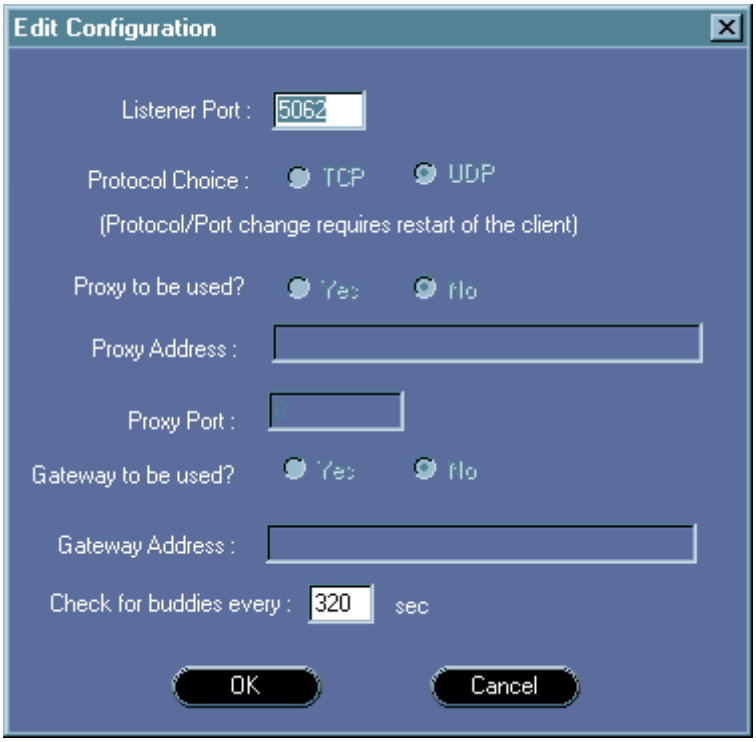
When the user clicks on this menu item, a dialog pops up with the first entry in the address book. The user can navigate the addressbook using the "Previous" and "Next" buttons. He can delete any entries by navigating to a particular entry and clicking on the "Delete" button. He can add a particular entry to the buddy list by checking the "Add to Buddy List" option. When he clicks on the "Done" button, it closes the pop up dialog.

ii) Add New Friend



Explained earlier *(*Online Buddy Tab*)*

iii) Configurations



When the user clicks on this menu item, a dialog pops up using which the user can configure the following

- The port on which he wants to receive the SIP signals
- The protocol to be used for exchanging SIP signals (This feature is not available in the evaluation release)
- If a proxy is used, the proxy's address and port number (This feature is not available in the evaluation release)
- If a gateway is used, the gateway's address (This feature is not available in the evaluation release)
- The frequency of spooling of buddies (for displaying the buddies who are online in the tab control)

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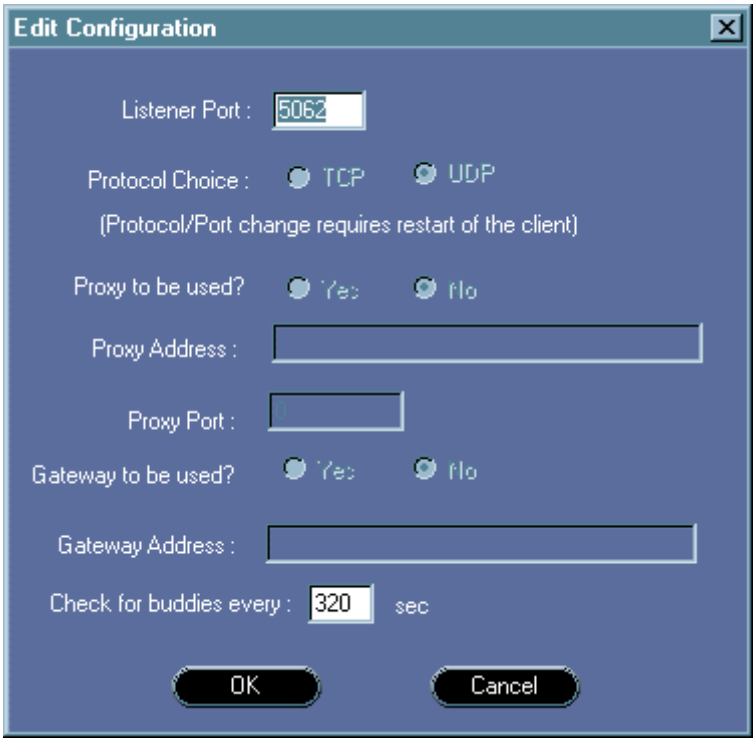
iii) Media Preferences



When this menu item is selected, a dialog pops up allowing the user to specify the codecs which he supports.

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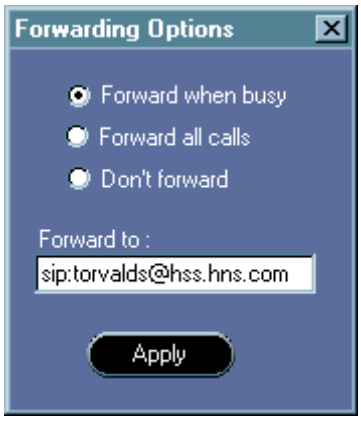
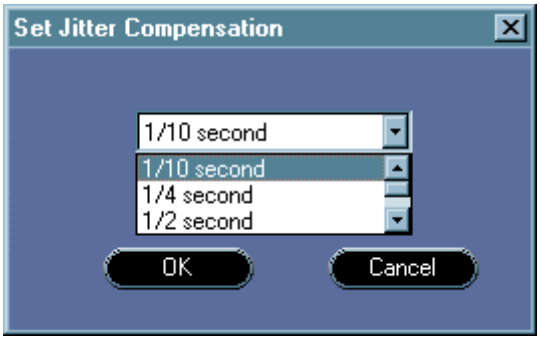
v) Forwarding



The user is provided a facility to forward incoming calls automatically to another SIP address. He can customize it further by checking one of the boxes for

- forwarding calls when busy(when the user has already established a call)
- forwarding calls at all times.

vi) Jitter Compensation



This option can be used to set the jitter compensation during conversation (audio).

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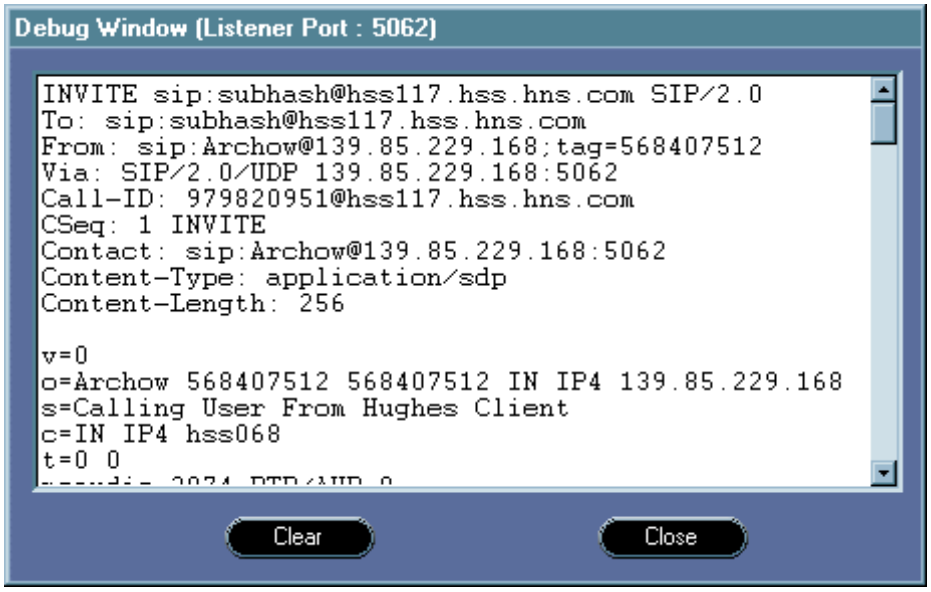
d) View

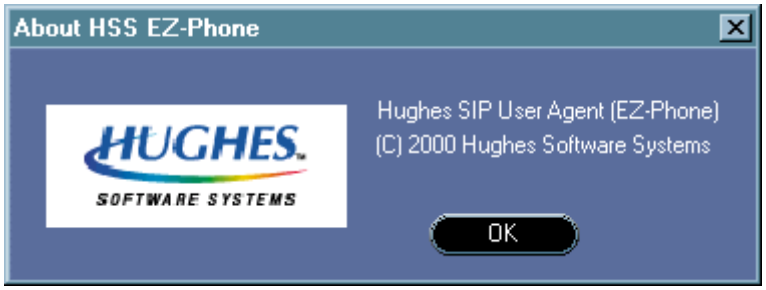
i) Debug Window



This allows the user to view the complete flow of the SIP signaling messages exchanged between the two or more parties through a dialog window that pops up. It has a "Clear" button that can be used to periodically clear the contents of the window. The dialog can be closed by clicking on the "Close" button.

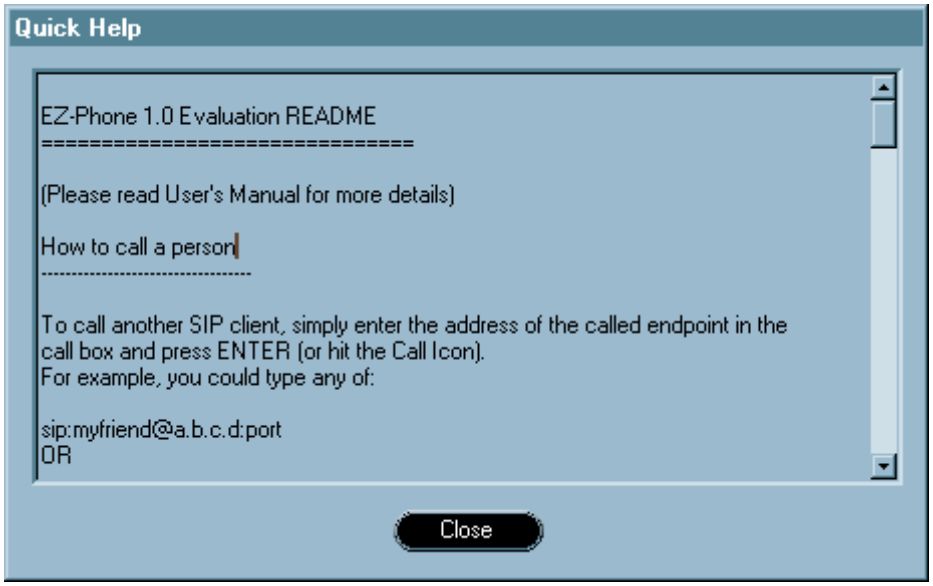
e) Help

i) About



This pops up a dialog with the details of the client.

ii) Quick Help


Quick Help
EZ-Phone 1.0 Evaluation README
==========
(Please read User's Manual for more details)
How to call a person

To call another SIP client, simply enter the address of the called endpoint in the call box and press ENTER (or hit the Call Icon).
For example, you could type any of:
sip:myfriend@a.b.c.d:port
OR
Close

This gives a brief description and quick tips for usage of the EZ-Phone.

6. Status messages

The following are the status messages that appear in the status bar during the operation of the EZ-Phone.

- ***Trying to call yourself!***: When the user tries to call himself (this is not allowed)
- ***Ready***: This indicates that the EZ-Phone is ready for use. It also indicates that the client is currently idle (not in any call) or has been reset.
- ***Ringing***: This indicates that the SIP client at the remote end is ringing (The remote user is being alerted)
- ***Calling Remote User***: This indicates that the remote SIP client is being invited to a call.
- ***Call Established*****:** This appears once a connection has been established.
- ***Attempting to answer call*****:** This appears when the user accepts an incoming call.
- ***Hung up***: This appears after a call has been terminated.
- ***Remote user is Busy***: This appears when the remote user is either busy or rejects a call.

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7. Error messages

The following are the error MessageBoxes that pop up during the operation of the SIP client.

Error Message Text	Reason
"Please enter a SIP URL"	This appears when the user attempts to call up without entering a SIP URL.
"The destination address should be of the form sip:[username]@<host>[:port]"	This appears when the user attempts to call up using an invalid (syntax) SIP URL.
"Unable to get remote client's host address."	This appears when the client is unable to resolve the remote hostname to a valid IP address.
Retransmissions Timed Out when trying to send <msg>. Terminating the call.	This occurs when the client is unable to send a SIP message to a remote client after 7 retransmissions.
Spool Time should be atleast 60 seconds.	This appears when a user tries to set the buddy list spool time to less than 60 seconds.
Please fill the SIP URL to forward calls to	This appears when the user tries to set forwarding options without specifying a URL.
Please select atleast one media type	This appears when the user does not select any media type from the Media Preferences dialog. Atleast one codec should be supported.
Cannot modify Jitter Compensation during a call.	This appears when the user tries to modify the Jitter compensation value during a call. Jitter compensation value can be modified only when the client is idle.
Please fill all the fields	This appears when the user leaves any of the fields in the addressbook blank while navigating the addressbook.

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